VoIP and Conventional PABX together based on Open Source Asterisk

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Abstraksi

The modern business activity can not avoid implementation of computer and communication. Cost and efficiency is an important factor as one of main consideration. Refer to the technology progress, convergence between computer and communication is very interesting. Refer to above situation, Gunadarma university considered to implement VoIP in the organization. There are 9 campus locations of Gunadarma university which utilized some old analog PABX. Currently, communication among locations among lectures and students utilize PSTN or mobile phone. The main problem is cost of communication and high mobility of person. Considerations of cost based on some factors such as budget of investment, cost of operational, budget of expandable, first implementation risk, operational risk. Currently, there are some products available in hardware and software level. Software products can be propiertery or open source solution. In depth consideration by looking for some factors were conducted. The result of consideration is to implement Asterisk open source software. In this paper, we will talk the background of VoIP in section one. Section two will discuss consideration of choosing Asterisk open source software and the design which is used for Gunadarma university based on current system by implementation Asterisk. Cost of communication is expensive Gunadarma has many campus location No of talk is high, there are 1,000 lectures and 25,000 students Need an economic, appropriate quality of solution for communication VoIP is a candidate, how to implement Asterisk open source to fullfill the demand.

Key words : *Asterisk,, Open Source, Proprietary, PSTN, VoIP*

1. Introduction

In the past, computer and communication worked separated where computer developed aimed at the ip-based network such as WAN and LAN that carried traditional data like email, browsing, ftp, and business applications, while communication developed aimed at the PSTN-based network that carried voice traffic. At that time, most people never thought about combining both of the technologies.

1.1 Shift From Traditional Voice Communication to VoIP

In the last decades, PSTN holds an important role in communication world as a platform to send voice traffic. It is undeniable that high cost of PSTN implementation becomes main problem for business. In the real application, PSTN is quite depending on an expensive hardware, maintenance/spare part, and system. Besides that, for the business community, fee that still be counted according to the distance has been felt burdening. It can not be imagined how much cost that must be spent if the business is quite big and it has an international scale that often needs a direct call among branch around the world.

Mixing the computer and communication network technology become a single network comes up as the solution for business. It is called as convergence. Since its appearance, people start to develop more creative ways to communicate. They start changing the basic function of computer which is only able to sending and receiving data to become a device that support voice communication. As well as on computer, communication device has been also changed in order supporting computer functions.

- The reduction of costs, because VoIP is relatively inexpensive.
- VoIP creates more services of phone system.
  that eliminates many problems.
- VoIP is mobile and flexible.

Besides the advantages, VoIP also had disadvantages that mostly people do not know. The disadvantage using VoIP include:

- There are some reliability problems.
- Voice over IP problem is the call quality.
- VoiP compared with traditional phone service, VoIP weakness is mostly the inability to make phone calls during an electrical outage or when the Internet connection is down for any reason [8].

So despite the advantages, enterprises find themselves locked into a single vendor solution, paying for licensing software, and trapped into whatever equipment, pricing, and patches the vendor issues. To address this problem, an open source solution is needed. Table 1 the comparison table between Proprietary VoIP Solution and Open Source VoIP Solution [1].

Organizations can deploy the VoIP solution in several ways without changing their network infrastructure or abandoning their investments in traditional PABX equipment. Here several scenarios which are implemented by Pingtel - Check Point Software Technologies Ltd [6].

1. Toll Bypass
2. Branch Office Extension
3. Mobile and Remote Workers

Table 1. VoIP Solution

	Proprietary VoIP Solution	**Open Source VoIP Solution**
Hardware	PABX application run on customer (or OEM) hardware/servers or are tied to vendor-specific routers and switches	Communications software runs on off-the-shell Linux servers and uses IO/TDM gateways from multiple vendors, driving down the cost of hardware
Software	PABX applications remain proprietary and costly	Communication software in open source model become virtually free
Service	An enterprise pays the chosen vendor for annual service and maintenance. No community of users of developers can be tapped into and the costumer has little influence on the future product road map	An enterprise pays for a lower-cost annual subscription service that covers all services an maintenance needs. An enterprise is able to speak directly to the developer, tap into the community, and exercise direct influence on future direction of the product.
Phone	Proprietary phones are sold at a premium to a captive audience.	Standards-based phones in many varieties become cost-effective because of competition.
Technology	Innovation is stifled by proprietary systems and protocols; vendor lock-in drives lack of interoperability and slows delivery of new features	Open source code base and SIP standard foster rapid innovation and assures universal interoperability
Market	Mature market with established players (about 4 percent CAGR)	New options for New option for costumers; new opportunities for many different vendors, adding to community.
Protocol	Proprietary protocol uses H.323, these include protocols that manage call setup and termination, negotiate channel usage, and handle authentication and security	Open source protocol uses SIP (Session Initiation Protocol), it is a newer, less complicated protocol that was designed specifically for VoIP.

2. METHODOLOGY AND ANALYSIS

2.1 Approach To Implement VoIP

Refer to the basic of VoIP, advantages and disadvantages, and also the examples of VoIP implementation, the next step we will discuss about approaching to implement VoIP.

When someone begins to implement VoIP, there are some decisions such as build VoIP from scratch by using open source-based, buy a ready-to-roll commercial version or opt for a middle course. The final selection hinges on the up-front cost as well as the time and resources devoted to building and configuring the system.

Selecting an approach to implement VoIP depends on business budget, technology expertise, willingness to get external support, risk, etc. If you have analyzed each approach like we have discussed above, you can make right decision to choose VoIP system which is fit for you [3]. As an example, after analyzing each approach, you get the result explain that you consider to reduce the implementation, operational cost, and you want get many benefits. You should have a better idea of how to begin your VoIP implementation by using open source VoIP like Asterisk.

2.2 Implementation of VoIP

As mentioned in abstract, Gunadarma University (GU) has 25,000 of students and 1000 lectures which is needed an economic and mobile communication solutions. Problems are occurred because of the communication technology that Gunadarma using now is costly and does not support students and lectures mobility. Based on the situation, Gunadarma tries looking for right solution and intends to switch their communication technology to VoIP. However, there are several factors that need to be considered before implementing it includes:

- Budget investment,
- Cost of operational,
- Budget of expandable,
- First implementation risk, and
- Operational risk.

Then we try to analyze both of Gunadarma main problems and each consideration. First, we see from the Gunadarma main problems, they have correlation with Gunadarma consideration to choose VoIP that is about cost and another consideration is about risk. We can conclude that the right VoIP solution for Gunadarma is choosing open source VoIP system and implementing the system by using existing network so that it is able to reduce both of the cost and risk of failure. One of open source VoIP that Gunadarma wants to implement is Asterisk. The reason that Gunadarma chooses Asterisk than other open source VoIP will be described below [2].

- Lower Cost.
- Flexible Features and Power Functions.
- Compatibility with a Wide Range of Platforms.
- Free and Abundant Support: The vibrant Asterisk user community provides no-cost support via the Internet.
- Continuity.
- Bandwidth

3. Design Of VoIP in Gunadarma University

3.1 Our current condition

Designing VoIP network is our next step after we choose Asterisk as our suitable VoIP system. Before we enter the stage, we must check first detail of our condition. There are two steps which cover the technical implementation
Network Assessment The purpose of network assessment is making sure the organization has the infrastructure in place to support the new system. A thorough network assessment can identify potential performance problems before we are knee-deep in installation [5].

- Network Inventory such as server, router, wireless access point, etc.
- Link Utilization
- In Table 2, we show bandwidth usage in Gunadarma University

Table 2. Link Utilization

Campus	Bandwidth	Peak (Kbps)	Average (Kbps)
A	10	384	128
C	10	384	128
D	1	2000	512
D(IM2)	10	1024	384
E	10	1024	384
G	10	512	128
H	10	128	32
J	10	256	128
K	10	256	64
TB. Simatupang	10	256	128
TB. Simatupang	1	768	512

- QoS Requirement
  We consider the minimal quality of service requirement.

Existing Phone Network
Figure 1 and Figure 2 show both of general and detail phone configuration in each campus before implementing Asterisk. The analog PABX only provided the standard telephone services and was not integrated into the data infrastructure.
How ever, each campus has been connected by wireless with bandwidth capacity for about 10Mbps and if we looked back Table 2, the bandwidth utilization each campus was below 1Mbps so it would quite support to develop Asterisk VoIP system.

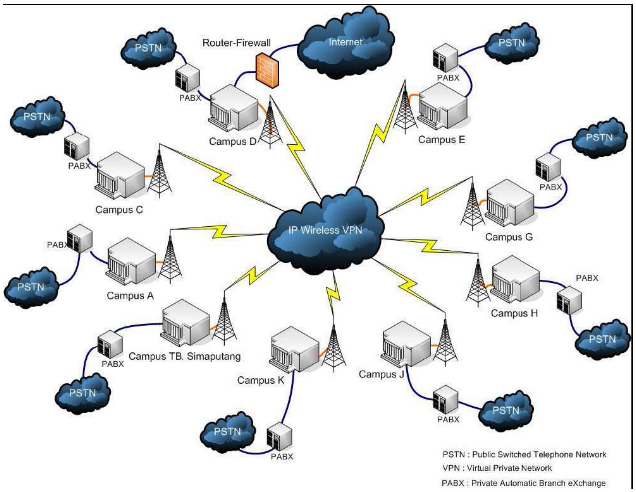

Fig. 1. General phone configuration in each campus before implementing Asterisk

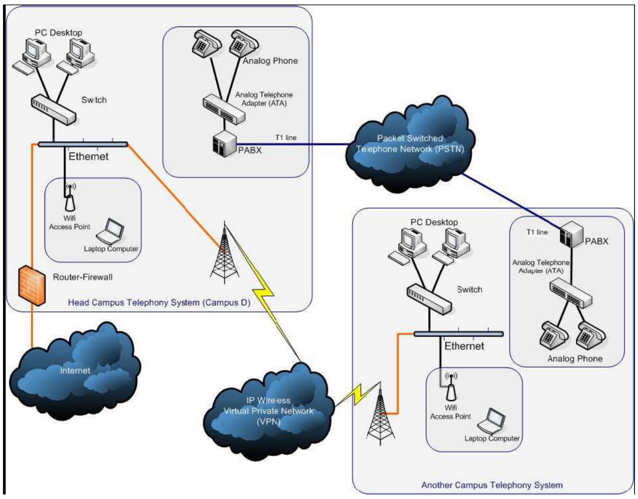

Fig. 2. Detail phone configuration in each campus before implementing Asterisk

3.2 Detail Design

In our VoIP design, we want to achieve several high-level objectives that the Gunadarma needs to understand before committing to a new voice technology. Our objectives included [7]:

- o Introducing a variety of converged communications devices into the enterprise
- o Understanding the benefits of an open standards based VoIP solution
- o Evaluating a phased migration plan from a traditional circuit-based PABX
- o to a packet-based PABX voice system
- o Measuring the cost savings and productivity impact of using VoIP, converged communications devices, and unified messaging

Requirement to build Asterisk When designing new Asterisk system, we must make right requirement that suitable with users need to reduce the risk of big hardware and software investment cost. Here are some questions and solutions as consideration before deciding if Asterisk is right for Gunadarma VoIP and Conventional PABX together based on Open Source Asterisk 7 [4].

1. How many users will Gunadarma new PABX need to support? Gunadarma has 9 campus included 25000 students and 1000 lectures and intends to implement centralized Asterisk that is going to be placed at Campus D, so we decide to use Asterisk (supports only 2-50 users) first as pilot project then we will choose upgrading to a business version (supports until 1000 users) or higher specification based on Asterisk pilot project result.
2. Do you have a high-performance server with a 3-GHz CPU, 1GB of RAM and a 100MB Ethernet interface? Gunadarma has a server which is match with the specification (supports about 50 to 100 concurrent callers). Our plan is to try to implement Asterisk on that server as pilot project. If we get satisfied result, we plan to upgrade to better specification in order to be able to fulfill users need. This kind of server costs about $2000.
3. What operating system will your server run? Asterisk was designed for Linux, but it also supports OpenBSD, FreeBSD, Mac OS X and Solaris. In our project, we implement Linux as operating system.
4. Will you be running VoIP, or digital or analog phone lines? Gunadarma implements VoIP by using the existing network. The objective is making convergence between old analog phone and new VoIP system.
5. Do you need to connect to a PTSN trunk line or connect ordinary telephones to the system? Gunadarma has planned using PSTN line as back up for Asterisk PABX system and for making calls to PSTN user which is not able to be covered by Gunadarma.
6. Do you want the easy route? If pressed for time, lacking the equipment, or users need it service as soon as possible, perhaps we may want to consider a packaged, managed service that provides most of what we need, such as Fonality's PABXtra package. But it is not our priority.
7. What is your budget? We have enough budget but we must spend it wisely. Since we intend to implement VoIP, we have made some priorities included :

    - o Reducing the cost implementation

- Buying right hardware for supporting this implementation
- Optimizing the current condition our budget estimation is about $2000 until $3000 for installing Asterisk system and buying some hardware that support Asterisk implementing like ip phone, single- and quad-span T1and E1 interfaces.

4. Results

Implementing of Design Next, design the new network architecture. Firstly, we worked with an up-to-date diagram of the existing network. Then, sketch out a new diagram incorporating the new VoIP gear. This blueprint will reveal any necessary upgrades or missing components, as well as any overlaps [5]. Figure 3 and figure 4 shows a network diagram of the general and detail configuration when implementing Asterisk. The Asterisk provided all the standard telephone services and was also tightly integrated into the messaging infrastructure.
This provided users with additional, enhanced VoIP applications like unified messaging. Voice traffic was prioritized and converged onto the standard production wide Area Network (WAN) with the data traffic. Voice trunks also remained connected to the public switched telephone network but now through Asterisk PABX, which acted as a gateway to the carrier. VoIP endpoints ultimately connected to the LAN. Existing ip phones connected to the LAN through the switch and existing analog phones connected to the LAN by using Analog Telephone Adapter (ATA). We implemented Wi-Fi handsets on our wireless network and will fully test and optimize them at a later date [7].

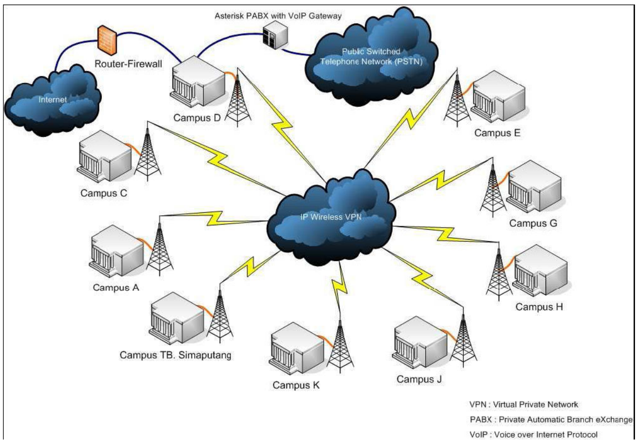

Fig. 3. General phone configuration in each campus when implementing Asterisk

5. Conclusion

Establishing an integrated communications platform can pay significant dividends in efficiency, productivity, and cost savings. To ensure that your business realizes the full potential of this powerful technology, follow a phased implementation plan. First, define specific communications objectives and assemble the necessary resources to implement the system. Next, analyze existing network architecture and determine how the legacy equipment can be integrated into the new IP-based system. Once this converged network–the backbone of your communications system–is in place, fine tune the system programming to address specific business needs. VoIP presents unique technical and logistical challenges at the outset but a successful implementation can streamline your business communications for years to come. The same condition happens to Gunadarma. By implementing VoIP especially open source-based VoIP like Asterisk, Gunadarma gets significant benefits mainly about cost savings. In addition, it drives the lectures and students mobility not only inside but also outside campus.

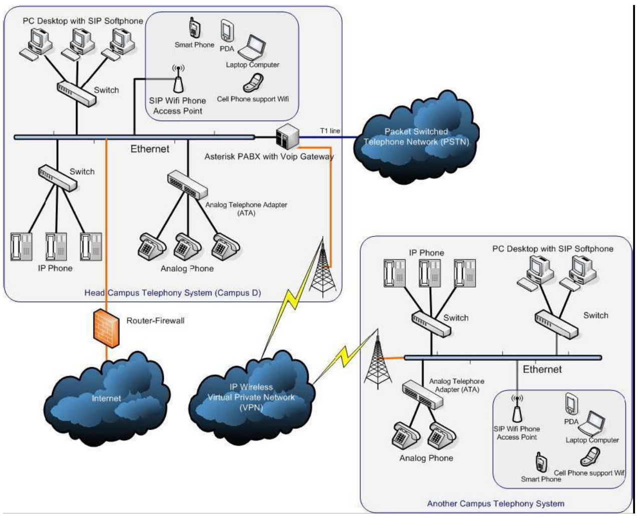

Fig. 4. Detail phone configuration in each campus when implementing Asterisk

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